outputs digital voice data in which the strength of said compressed voice code inputted immediately prior to said dummy signal is reduced.

Claim 4 (twice amended). A voice transceiver according to claim 1 [characterized in] further comprising:

a microphone means for inputting voice data;

a second conversion means for converting said voice data into a digital signal, and outputting this conversion result as other digital voice data; and

an echo component removal means for removing the echo component contained in said other digital voice data.

REMARKS

Claims 1, 3, and 4 have been amended. Claims 1-4 remain in the application.

In accordance with the Examiner's requirement, the title has been amended to read as follows: VOICE TRANSCEIVER WHICH ELIMINATES UNDERFLOW AND OVERFLOW FROM THE SPEAKER OUTPUT BUFFER. If an alternative title is preferred by the Examiner, he is requested to supply a suitable alternative.

Page 1 of the specification has been amended to delete reference to a Japanese Patent Application being incorporated by reference. Priority to this application has already been claimed in the declaration and a certified priority document has been supplied to the U.S. Patent and Trademark Office.

Pages 2 and 3 of the specification has been amended to comply with the reference numerals set forth in Figure 3 of the application.

The claimed invention

As explained on page 3 of the application, in conventional system, if the digital voice input signal is not smoothly and regularly supplied, there is a severe

degradation in sound quality emanating from speaker 801. This degradation can happen with personal computer systems when processing the voice transceiver as software, or in multimedia terminals where image data is mixed with the voice codes and transmitted therewith (resulting in degradation of the quality of voice reception even in the voice transmission processing portion of a multimedia transmission terminal). Page 4 of the application indicates that enlargement of the SP output buffer has been attempted to avoid degradation; however, this caused an increase in the shift distance over which the digital voice input signal must pass from the point of input to the point of output.

With reference to Figure 1 of the patent application, it can be seen that a selective disposal unit 200 selectively discards digital voice data from the voice decoder to the SP output buffer, and that an insertion/disposal control unit 100 monitors the remaining data amount of the digital voice data stored in the SP output buffer 501. As noted on page 8 of the application, at lines 18-20, the insertion/disposal control unit 100 outputs both a dummy compression code to the decoding code buffer 301, and a discard request signal to the selective disposal unit 200.

In operation, as discussed at page 9 of the application at lines 11 et seq., if the selective disposal unit 200 has not received a discard request from control unit 100, then the supplied digital voice data is written into the SP output buffer 501. However, if a discard request is received, then the corresponding digital voice data is discarded rather than written to the SP output buffer. The control unit 100 monitors the remaining data amount of the digital voice data stored in the output buffer 100. When this amount falls below a previously set first threshold value, a dummy voice code is output to the decoding code buffer 301. This first threshold represents the lower limit of the data amount stored in the SP output buffer 501 at which interruption of the output voice does not occur in the speaker 801. Thereafter, the supply of dummy voice code serves as a trigger by means of which the operation of generating the SP output voice is started. As noted on page 10 of the application, the voice data

accumulates in the SP output buffer 501 until the data amount stored in the buffer exceeds the first threshold value. Conversely, when the remaining data amount in the SP output buffer exceeds a second threshold value, a discard request is issued to the disposal unit 200, whereupon discard processing of the digital voice data is performed and the supply of voice data is halted until the data amount returns to a level below the second threshold value (i.e., the upper limit of the data amount which is to be stored in the SP output buffer 501).

Figure 2 shows a second embodiment of the invention similar to that shown in Figure 1. It is noted that a reference input signal buffer 901 receives data from the selective disposal unit 200, and is connected to an acoustic echo canceller 902. In the second embodiment, the selective disposal unit operates in the same manner as described above with respect to the discard request signal (see page 14 of the application) and with respect to dummy voice codes (see page 15 of the application).

The effect of both embodiments is that the speaker unit is able to smoothly output an output voice without breakup.

The acoustic echo canceller 902 references the digital voice data in the buffer 901, and suppresses the echo component in the digital input signal to be supplied to the voice encoder 402. As explained on page 16 of the patent application, the acoustic echo canceller provides the additional effect of stability since the contents of the output voice of the speaker 801 and the reference input signal buffer 901 are always in agreement by means of the insertion/disposal control unit 100.

Claim 1 has been amended to remove the words "characterized in" from the preamble. The scope of the claim has not been changed. Rather, the amendment makes the claim language more appropriate for U.S. practice.

An important feature of claim 1 not found in the prior art is <u>the detection means for detecting the quantity of data in said digital voice data stored in said buffer, and outputting a detection signal as a detection result</u>. As discussed above in conjunction with both Figures 1 and 2 of the invention, the amount of data in the SP output buffer is monitored (either by the insertion/disposal control unit

directly–Figure 1, or through the reference input signal buffer 901–Figure 2), and a detection signal is then output from the detection means, and the conversion means converts the digital signal to an analog signal <u>based on the detection signal</u> (as specified in claim 1). See page 16, lines 18 et seq., of the patent application for support for the "detection means". As explained above, if the digital signal is a discard request, then the corresponding digital voice data is discarded rather than written to the SP output buffer, but if a discard request is not received, then the supplied digital voice data is written to the buffer.

Claim 2 requires a <u>data control means</u> for controlling the output of digital voice data from the converter based on the detection signal, wherein the data control means outputs <u>a dummy code</u> to the expansion means (i.e, element 301) when the amount of digital voice data in the buffer is less than an amount required for play back, but in cases where the buffer is approaching overflow, the control means prevents output of the digital voice data to the conversion means.

Claim 3 is related to the expansion means outputting digital voice data in which the strength of the compressed voice code immediately prior to the dummy signal is reduced. See the paragraph bridging pages 17 and 18 of the patent application.

Claim 4 is directed to the embodiment shown in Figure 2 of the patent application.

<u>35 U.S.C. 112, second paragraph</u>

Claims 3 and 4 were rejected under

Claim 3 now depends from claim 2 of the application. Claim 2 provides antecedent basis for "said dummy code".

Claim 4 is directed to the embodiment shown in Figure 2 of the patent application. The Examiner has not specifically identified any particular problem with the language used in claim 4. For the Examiner's reference, the microphone means

corresponds to 802, the second conversion means corresponds to 602, and the echo component removal means corresponds to 902. Claim 4 has been amended by canceling "characterized in" from the preamble for formal reasons; however, the scope of the claim is the same as original filed.

35 U.S.C. 102 and 35 U.S.C. 103

Claims 1-3 were rejected as being anticipated by U.S. Patent 5,526,353 to Henley, and claim 4 was rejected as being obvious over Henley in view of U.S. Patent 5,617,423 to Li. The Applicant traverses these rejections.

In particular, the Examiner relies upon column 14, lines 10-55 as teaching the "detection means...detection result", and upon column 7, lines 27-33 as teaching the "conversion means...said detection signal...and speaker means...".

Henley describes a system for communicating audio data in a packet based computer network. There is a packet assembly circuit for constructing a data packet from a portion of a stream of digital audio data. This circuit generates a position identifier that indicates a temporal position of the portion relative to the stream, and this position identifier is inserted into the data packet for queuing for transmission through the computer network. In Henley, there is also a packet disassembly circuit for receiving a data packet and inserting the portion into an absolute location in the buffer, the position identifier determining the location (see column 16, lines 30-49). This allows synchronization with adjacent portion of the stream of digital audio data in the buffer to compensate for the variable periods of transmission time.

Henley does not disclose any element that is akin to the "detection means...detection result", "conversion means...said detection signal", as suggested by the Examiner. In the claimed invention, the detection means detects the quantity of data in said digital voice stored in said buffer, and the conversion means converts the digital voice data into analog voice data based on said detection signal.. That is, the claimed invention is drawn to a system whereby the overfill and underfill state of the buffer is monitored and adjusted. Claim 2 is directed to using dummy codes to

compensate for underfill.

Column 14, lines 10-55 of Henley, which was referenced by the Examiner, is drawn to placement in the "receiving buffer 510 as a function of the position identifier 370 contained in each data packet" (see column 14, lines 13-14). That is, "The position identifier 370 directs each audio sample 380 into specified absolute positions of the receiving buffer 510 at the receiving end". Hence, in Henley there is no "detection means for detecting the quantity of data in said digital voice data stored in said buffer" as required in claim 1. Rather, Henley contemplates a system for assigning positions within a buffer, and is not concerned with underfill or overfill of a speaker output buffer that directs a signal to converter.

The Examiner has referenced column 15, lines 7-46, as being illustrative of a control means as specified in claim 2 of the application. This is not correct. The claimed control means outputs a dummy code to said expansion means when the digital voice data stored in the buffer means is less than an amount required for playback, and prevents the output of digital voice data to the conversion means when the buffer means approaches an overflow amount. With reference to Henley, at column 15, it should be noted that Table 1 in column 14 is being referenced. Samples 5 and 7 are dealt with in Column 15 of Henley. Note that both samples are provided with a position identifier as discussed above, which precisely locates the sample within a buffer (the quantity of data in said digital voice data stored in said buffer is not being detected or used to enable or disenable transmission to the conversion means). Sample 5 is short; therefore, white noise is added at the location of Sample 5. This is not akin to a controller providing a dummy code to said expansion means when the digital voice data stored in the buffer means is less than an amount required for playback. Rather, this is an operation performed in Henley for the data packet assembly operation. Sample 7 is long, therefore 2 bytes are removed to achieve a preset size. Again, this is not akin to said data control means does not allow the output of said digital voice data to said conversion means. Rather, again, this is an operation performed in Henley for data packet assembly operation.

Reference made by the Examiner to column 7, lines 27-33 of Henley does not anticipate or make obvious the claimed invention. This passage in Henley does not discuss operation of the conversion means based on said detection signal. Furthermore, this passage refers to a compression/analog conversion circuit. In sharp contrast, the detection means in the claimed invention digital voice data in the buffer that is already expanded by said expansion means.

With respect to a combination of Li and Henley, it is noted that Li, like Henley, lacks the claimed detection means and the claimed conversion means which provides for conversion to analog voice data based on the detection signal. As such, no combination of Li and Henley would make claim 1 obvious, or claim 4 obvious which depends from claim 1.

With respect to the echo component removal means (element 902 in Figure 2 of the patent application), it is noted that Li does not contemplate receiving a signal that is identical to that output to a speaker output buffer. As noted above, the echo removal component assures that the output voice of the actual speaker and those of the buffer means are always in agreement. Claim 4 requires that the echo component removal means removes the echo component in said other digital voice data (this comes from microphone 802). In contrast, Li is directed to echo cancellation to avoid feedback between transmitted and received signals. In Li, there is no concept of a reference input signal buffer 901 having identical information as that passed to speaker output buffer 501, and using this information for echo cancellation.

In view of the above, the application should now be in condition for allowance. Reconsideration and allowance of claims 1-4 at an early date is requested.

Respectfully submitted,

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